

Configuring E.164 in AOS Voice Devices

E.164 is an International Telecommunications Union (ITU-T) recommendation for standardized dialing throughout the world. Each country and territory has a uniquely assigned country code (CC) of 1-3 numbers. For example, the CC for the United States and Canada is “1” while the United Kingdom and Northern Ireland is “44.” The typical structure for an E.164 call is as follows:

Country Code + Nationally Significant Number
(1-3 digits) (Max 15 digits)

E.164 can be configured in AOS voice devices, and these numbers appear with a + prefix. *Note that variations of the E.164 structure are allowed in the ADTRAN. For example, a typical E.164 number for a call inside the U.S. may look like this: +12565559001. However, the country code and/or area code may be removed to accommodate network requirements.* The ADTRAN AOS voice device can be configured in two ways to make calls with the E.164 format. The following guide will provide all the necessary commands for this configuration.

E.164 Dialing when Specifying Prefix and Country Code

The commands needed for this mode are global commands:

voice country-code 1
voice international-prefix abbreviated

The CC is set to “1” by default, which is the country code for the United States. Setting international prefix to “abbreviated” actually turns on the E.164 format. These are the only commands that are needed in order to enable E.164 formatting when dialing out from behind the AOS voice device with a “011” before the called party number. “011” is the international prefix used to dial somewhere outside of the United States.

Someone dialing from a user (SIP, Analog) or trunk (PRI, CAS T1) behind the AOS voice device will need to specify the international prefix, “011,” and country code, before dialing the number. In the following debug, the analog voice user dials “256-555-9070” after specifying the prefix (011) and country code (1). The debugs used to capture this call are:

debug voice toneservices
debug sip stack messages

```
15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (0) event
15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: stopping
15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: TDM unmap
```

```

15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: release
15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (1) event
15:42:47 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (1) event
15:42:48 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (1) event
15:42:48 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (2) event
15:42:48 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
15:42:49 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (6) event
15:42:49 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
15:42:50 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
15:42:50 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
15:42:50 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (9) event
15:42:50 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (0) event
15:42:51 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (7) event
15:42:51 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (0) event
15:42:55 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: stopping
15:42:55 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: TDM unmap
15:42:55 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: release

15:42:55 SIP.STACK MSG Tx: UDP src=10.19.213.121:5060 dst=10.1.4.5:5060
15:42:55 SIP.STACK MSG INVITE sip:+12565559070@10.1.4.5:5060 SIP/2.0
15:42:55 SIP.STACK MSG From: "First Last"<sip:5559076@10.1.4.5:5060;transport=UDP>;tag=2fa1c28-a13d579-13c4-58d94-5951f522-58d94
15:42:55 SIP.STACK MSG To: <sip:+12565559070@10.1.4.5:5060;transport=UDP>
15:42:55 SIP.STACK MSG Call-ID: 2fe5eb0-a13d579-13c4-58d94-1b803f46-58d94@10.1.4.5
15:42:55 SIP.STACK MSG CSeq: 1 INVITE
15:42:55 SIP.STACK MSG Via: SIP/2.0/UDP 10.19.213.121:5060;branch=z9hG4bK-58d94-15b10b96-5bc4facc
15:42:55 SIP.STACK MSG Max-Forwards: 70
15:42:55 SIP.STACK MSG Supported: 100rel,replaces
15:42:55 SIP.STACK MSG Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK, REFER, REGISTER
15:42:55 SIP.STACK MSG User-Agent: ADTRAN_Total_Access_916e/16.03.00.E
15:42:55 SIP.STACK MSG Contact: <sip:5559076@10.19.213.121:5060;transport=UDP>
15:42:55 SIP.STACK MSG Content-Type: application/SDP
15:42:55 SIP.STACK MSG Content-Length: 210
15:42:55 SIP.STACK MSG
15:42:55 SIP.STACK MSG v=0
15:42:55 SIP.STACK MSG o=- 1204580575 1204580575 IN IP4 10.19.213.121
15:42:55 SIP.STACK MSG s=-
15:42:55 SIP.STACK MSG c=IN IP4 10.19.213.121
15:42:55 SIP.STACK MSG t=0 0
15:42:55 SIP.STACK MSG m=audio 10048 RTP/AVP 0 18
15:42:55 SIP.STACK MSG a=rtpmap:0 PCMU/8000
15:42:55 SIP.STACK MSG a=rtpmap:18 G729/8000
15:42:55 SIP.STACK MSG a=fmtp:18 annexb=no
15:42:55 SIP.STACK MSG a=silenceSupp:off - - - -

```

The *voice toneservices* debug catches the dialed digits, *011-1-256-555-9070*. The SIP debug catches the SIP messages. The INVITE URI and the To header show the AOS voice device formatting the dialed number to meet the E.164 standard. *01112565559070* becomes *+12565559070*.

E.164 Dialing without Specifying Prefix and Country Code

E.164 formatting can be accomplished without the end user ever having to dial an international prefix or country code. By using the match/sub commands in the AOS voice device, any dialed number that matches a specified pattern can be substituted with the international prefix and country code set in the unit.

The commands needed for this mode are the same global commands as mentioned earlier:

voice country-code 1
voice international-prefix abbreviated

In addition, the combination of appropriate match/sub commands needs to be applied on the SIP trunk. Below are a few examples:

match 1-NXX-NXX-XXXX sub 011-1-NXX-NXX-XXXX
match NXX-NXX-XXXX sub 011-1-NXX-NXX-XXXX
match NXX-XXXX sub 011-1-256-NXX-XXXX

In the following debug, the analog voice user dials 2565559070, which would match the middle pattern above. The AOS voice device will do the substitution automatically, so 2565559070 becomes 01112565559070 internally. Since the international prefix is set to abbreviated, the final number appears as +12565559070. The debugs used to capture this call are:

debug voice toneservices
debug voice summary
debug sip stack messages

```
09:44:20 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (2) event
09:44:20 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: stopping
09:44:20 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: TDM unmap
09:44:20 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - DialTone Generation: release
09:44:21 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
09:44:21 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (6) event
09:44:22 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
09:44:22 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
09:44:22 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (5) event
09:44:23 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (9) event
09:44:23 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (0) event
09:44:23 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (7) event
09:44:24 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: received digit (0) event
09:44:28 VOICE.SUMMARY voice user 9076 cos allowed the call to Extensions
09:44:28 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: stopping
09:44:28 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: TDM unmap
09:44:28 TONESERVICES.EVENTS fxs 0/1 - dsp 0/2.1 - Tone Detection: release
09:44:28 VOICE.SUMMARY 9076 is calling T01 (2565559070).
09:44:28 VOICE.SUMMARY DNIS Substitution made - Original dialed #(2565559070) New dialed #(01112565559070)

09:44:28 SIP.STACK MSG Tx: UDP src=10.19.213.121:5060 dst=10.1.4.5:5060
09:44:28 SIP.STACK MSG INVITE sip:+12565559070@10.1.4.5:5060 SIP/2.0
09:44:28 SIP.STACK MSG From: "First Last"<sip555:9076@10.1.4.5:5060;transport=UDP>;tag=2fc5b80-a13d579-13c4-68b11-44d8e5ea-68b11
09:44:28 SIP.STACK MSG To: <sip:+12565559070@10.1.4.5:5060;transport=UDP>
09:44:28 SIP.STACK MSG Call-ID: 2fe6480-a13d579-13c4-68b11-69a131ee-68b11@10.1.4.5
09:44:28 SIP.STACK MSG CSeq: 1 INVITE
09:44:28 SIP.STACK MSG Via: SIP/2.0/UDP 10.19.213.121:5060;branch=z9hG4bK-68b11-198f3b46-44466ff4
09:44:28 SIP.STACK MSG Max-Forwards: 70
09:44:28 SIP.STACK MSG Supported: 100rel,replaces
09:44:28 SIP.STACK MSG Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK, REFER, REGISTER
09:44:28 SIP.STACK MSG User-Agent: ADTRAN_Total_Access_916e/16.03.00.E
09:44:28 SIP.STACK MSG Contact: <sip:5559076@10.19.213.121:5060;transport=UDP>
09:44:28 SIP.STACK MSG Content-Type: application/SDP
09:44:28 SIP.STACK MSG Content-Length: 210
09:44:28 SIP.STACK MSG v=0
09:44:28 SIP.STACK MSG o=- 1204645468 1204645468 IN IP4 10.19.213.121
09:44:28 SIP.STACK MSG s=-
09:44:28 SIP.STACK MSG c=IN IP4 10.19.213.121
```

```
09:44:28 SIP.STACK MSG      t=0 0
09:44:28 SIP.STACK MSG      m=audio 10048 RTP/AVP 0 18
09:44:28 SIP.STACK MSG      a=rtpmap:0 PCMU/8000
09:44:28 SIP.STACK MSG      a=rtpmap:18 G729/8000
09:44:28 SIP.STACK MSG      a=fmtp:18 annexb=no
09:44:28 SIP.STACK MSG      a=silenceSupp:off - - - -
```

The *voice toneservices* debug catches the dialed digits, 256-555-9070. The *voice summary* debug shows the DNS substitution, 256-555-9070→011-1-256-555-9070. The SIP debug catches the SIP messages. The INVITE URI and the To header show the AOS voice device formatting the dialed number to meet the E.164 standard. 2565559070 becomes +12565559070.

Inbound E.164 Calls to the AOS Voice Device

When an inbound call in the E.164 format hits the SIP trunk on the AOS voice device, the switchboard will strip the + and the country code from the number in the Request-URI. There must be a sip-identity defined on a voice user (if the voice user number differs from the number being received) or accept template on a grouped-trunk that matches the number (without the + or the country code) for the call to be routed. Here is an example config for a voice user and voice trunk in an AOS voice device.

```
voice user 5559001
connect fxs 0/1
password "1234"
sip-identity T01 2565559001
```

```
voice grouped-trunk PRI
description "PRI to PBX"
trunk T02
accept 2565559001 cost 0
```

If the from user is also being presented in E.164 format, the following global configuration command must be configured to avoid the E.164 format calling party information from being presented to the destination user or trunk:

```
(config)#ip sip grammar from user international
```